Claims

1. (original) A method for lossless compression of at least a portion of an audio signal, the method comprising:

for a sample currently being encoded in the portion of the audio signal, processing a set of other samples using an adaptive filter to predict a value for the sample;

producing a prediction residue for the current sample;

updating filter coefficients of the adaptive filter;

detecting whether the current sample is located about a transient in the audio signal; and

varying an adaptation rate for said updating the adaptive filter coefficients according to a result of said detecting.

2. (currently amended) The method of claim 1 wherein said varying the ~~update speed~~ adaptation rate increases the adaptation rate where the current sample is detected to be located about the audio signal transient.

3. (currently amended) A method for lossless compression of at least a portion of a multi-channel audio signal, the method comprising:

processing a set of samples of the multi-channel audio signal using an adaptive filter to predict a value for a current sample in a current channel of the audio signal currently being encoded, wherein the set of samples comprises samples in other channels of the audio signal;

producing based on the adaptive filter processing a prediction residue for the current sample;

updating filter coefficients of the adaptive filter;

detecting whether the current sample is located about a transient in the audio signal; and

varying an adaptation rate for said updating the adaptive filter coefficients according to a result of said detecting; and

encoding the value of the current sample based on the prediction residue, ~~whereby said adaptive filter processing based also on samples in other channels reduces~~ thereby reducing inter-channel redundancy of the audio signal.

4. (original) The method of claim 3 wherein the adaptive filter is a least mean square filter.

5. (currently amended) A method for lossless compression of at least a portion of an audio signal, the method comprising:

for a sample currently being encoded in the portion of the audio signal, producing a prediction residue using an adaptive filter; ~~and~~

encoding the prediction residue using Golomb coding;

updating filter coefficients of the adaptive filter;

detecting whether the current sample is located about a transient in the audio signal; and

varying an adaptation rate for said updating the adaptive filter coefficients according to a result of said detecting.

6. (original) The method of claim 5 wherein the Golomb coding has a divisor not equal to a power of 2.

7. (original) The method of claim 5 wherein the divisor is 3.

8. (original) An audio decoder for processing a compressed data stream encoded via the method of any one of claims 1 through 6 to produce an audio signal substantially corresponding to the original input signal.

9. (original) A computer readable medium having a program carried thereon executable on a computer to perform a method for lossless compression of at least a portion of an audio signal, the method comprising:

for a sample currently being encoded in the portion of the audio signal, processing a set of other samples using an adaptive filter to predict a value for the sample;

producing a prediction residue for the current sample;

updating filter coefficients of the adaptive filter;

detecting whether the current sample is located about a transient in the audio signal; and

varying an adaptation rate for said updating the adaptive filter coefficients according to a result of said detecting.

10. (currently amended) The computer readable medium of claim 9 wherein said varying the ~~update speed~~ adaptation rate increases the adaptation rate where the current sample is detected to be located about the audio signal transient.

11. (currently amended) A computer readable medium having a program carried thereon executable on a computer to perform a method for lossless compression of at least a portion of a multi-channel audio signal, the method comprising:

processing a set of samples of the multi-channel audio signal using an adaptive filter to predict a value for a current sample in a current channel of the audio signal currently being encoded, wherein the set of samples comprises samples in other channels of the audio signal;

producing, based on the adaptive filter processing, a prediction residue for the current sample;

updating filter coefficients of the adaptive filter;

detecting whether the current sample is located about a transient in the audio signal; and

varying an adaptation rate for said updating the adaptive filter coefficients according to a result of said detecting; and

encoding the value of the current sample based on the prediction residue, ~~whereby said adaptive filter processing based also on samples in other channels reduces~~ thereby reducing inter-channel redundancy of the audio signal.

12. (original) The computer readable medium of claim 11 wherein the adaptive filter is a least mean square filter.

13. (currently amended) A computer readable medium having a program carried thereon executable on a computer to perform a method for lossless compression of at least a portion of an audio signal, the method comprising:

for a sample currently being encoded in the portion of the audio signal, producing a prediction residue using an adaptive filter; ~~and~~

encoding the prediction residue using Golomb coding;

updating filter coefficients of the adaptive filter;

detecting whether the current sample is located about a transient in the audio signal; and

varying an adaptation rate for said updating the adaptive filter coefficients according to a result of said detecting.

14. (original) The computer readable medium of claim 13, wherein the Golomb coding has a divisor not equal to a power of 2.

15. (original) The computer readable medium of claim 13 wherein the divisor is 3.

16. (original) An audio encoder for losslessly compressing at least a portion of an audio signal, the audio encoder comprising:

an adaptive filter operating, for a sample currently being encoded in the portion of the audio signal, to process a set of other samples to produce a prediction residue for the current sample;

the adaptive filter further updating filter coefficients based on the processing the set of other samples according to an adaptation rate;

a transient detector for detecting a transient has occurred located about the current sample in the audio signal; and

an adaptation rate controller for varying an adaptation rate of the adaptive filter responsive to the transient detector.

17. (original) The audio encoder of claim 16 wherein said varying the adaptation rate increases the adaptation rate when a transient is detected by the transient detector.

18. (currently amended) A multi-channel audio encoder for lossless compression of at least a portion of a multi-channel audio signal, the method comprising:

an adaptive filter for processing a set of samples of the multi-channel audio signal using an adaptive filter to predict a value for a current sample in a current channel of the audio signal currently being encoded, wherein the set of samples comprises samples in other channels of the

audio signal, and producing based on the processing a prediction residue for the current sample; ~~and~~

an entropy encoder for encoding the value of the current sample based on the prediction residue, whereby said adaptive filter processing based also on samples in other channels reduces inter-channel redundancy of the audio signal;

the adaptive filter further updating filter coefficients based on the processing the set of samples according to an adaptation rate;

a transient detector for detecting a transient has occurred located about the current sample in the audio signal; and

an adaptation rate controller for varying an adaptation rate of the adaptive filter responsive to the transient detector.

19. (original) The multi-channel audio encoder of claim 18 wherein the adaptive filter is a least mean square filter.

20. (currently amended) An audio encoder for lossless compression of at least a portion of an audio signal, the audio encoder comprising:

an adaptive filter for producing a prediction residue for a sample currently being encoded in the portion of the audio signal; ~~and~~

a Golomb coder for encoding the prediction residue using Golomb coding[[,]];

the adaptive filter further updating filter coefficients;

a transient detector for detecting a transient has occurred located about the sample currently being encoded in the portion of the audio signal; and

an adaptation rate controller for varying an adaptation rate of the adaptive filter responsive to the transient detector.

21. (original) The audio encoder of claim 20 wherein the Golomb coding has a divisor not equal to a power of 2.

22. (original) The audio encoder of claim 20 wherein the divisor is 3.